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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/468,138	12/21/1999	STEPHEN DOUGLAS PETERS	85773-161	3272

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SMART & BIGGAR  
1000 DE LA GAUCHETIERE STREET WEST  
SUITE  
MONTREAL, H3B4W5  
CANADA

EXAMINER

HARPER, VINCENT P

ART UNIT	PAPER NUMBER
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2654

DATE MAILED: 10/28/2002

Please find below and/or attached an Office communication concerning this application or proceeding.

# Office Action Summary

Application No.

09/468,138

Applicant(s)

PETERS ET AL.

Examiner

V. Paul Harper

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-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

## Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133).
- Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

## Status

- 1) ☐ Responsive to communication(s) filed on \_\_\_\_.
- 2a) ☐ This action is FINAL. 2b) ☒ This action is non-final.
- 3) ☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

## Disposition of Claims

- 4) ☒ Claim(s) 1-36 is/are pending in the application.
- 4a) Of the above claim(s) \_\_\_\_ is/are withdrawn from consideration.
- 5) ☐ Claim(s) \_\_\_\_ is/are allowed.
- 6) ☒ Claim(s) 1-36 is/are rejected.
- 7) ☐ Claim(s) \_\_\_\_ is/are objected to.
- 8) ☐ Claim(s) \_\_\_\_ are subject to restriction and/or election requirement.

## Application Papers

- 9) ☐ The specification is objected to by the Examiner.
- 10) ☐ The drawing(s) filed on \_\_\_\_ is/are: a) ☐ accepted or b) ☐ objected to by the Examiner.
- Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).
- 11) ☐ The proposed drawing correction filed on \_\_\_\_ is: a) ☐ approved b) ☐ disapproved by the Examiner.
- If approved, corrected drawings are required in reply to this Office action.
- 12) ☐ The oath or declaration is objected to by the Examiner.

## Priority under 35 U.S.C. §§ 119 and 120

- 13) ☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).
- a) ☐ All b) ☐ Some \* c) ☐ None of:
1. ☐ Certified copies of the priority documents have been received.
2. ☐ Certified copies of the priority documents have been received in Application No. \_\_\_\_.
3. ☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).
- \* See the attached detailed Office action for a list of the certified copies not received.
- 14) ☐ Acknowledgment is made of a claim for domestic priority under 35 U.S.C. § 119(e) (to a provisional application).
- a) ☐ The translation of the foreign language provisional application has been received.
- 15) ☐ Acknowledgment is made of a claim for domestic priority under 35 U.S.C. §§ 120 and/or 121.

## Attachment(s)

- 1) ☒ Notice of References Cited (PTO-892)
- 2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
- 3) ☒ Information Disclosure Statement(s) (PTO-1449) Paper No(s) 2.
- 4) ☐ Interview Summary (PTO-413) Paper No(s) \_\_\_\_.
- 5) ☐ Notice of Informal Patent Application (PTO-152)
- 6) ☐ Other: \_\_\_\_\_

## DETAILED ACTION

### *Information Disclosure Statement*

1. The Examiner has considered the references listed in the Information Disclosure Statement dated 12/21/99. A copy of the Information Disclosure Statement is attached to this office action.

### *Claim Rejections - 35 USC § 102*

The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

A person shall be entitled to a patent unless –

(e) the invention was described in a patent granted on an application for patent by another filed in the United States before the invention thereof by the applicant for patent, or on an international application by another who has fulfilled the requirements of paragraphs (1), (2), and (4) of section 371(c) of this title before the invention thereof by the applicant for patent.

The changes made to 35 U.S.C. 102(e) by the American Inventors Protection Act of 1999 (AIPA) do not apply to the examination of this application as the application being examined was not (1) filed on or after November 29, 2000, or (2) voluntarily published under 35 U.S.C. 122(b). Therefore, this application is examined under 35 U.S.C. 102(e) prior to the amendment by the AIPA (pre-AIPA 35 U.S.C. 102(e)).

2. Claims 1-36 are rejected under 35 U.S.C. 102(e) as being anticipated by Kuhn et al., (U.S. Patent 6,327,565), hereinafter referred to as Kuhn.

Regarding claims 1, 7 and 35, Kuhn discloses a system for speaker adaptation that includes the following: a new speaker input (col. 5, lines 26-39, Fig. 3, 40), which

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corresponds to "an input for receiving an input signal derived from a spoken utterance that contains at least one speech element that potentially matches the given speech element"; a set of HMMs 44 (one for each sound) (col. 5, lines 23-34), which corresponds to "a model group associated to the given speech element, said model group comprising a plurality of speech models, each speech model of said plurality of speech models being a different representation of the given speech element"; with an inherent processing unit to generate an adapted model based in the input (Fig. 3 52 col. 5, lines 50-57), which corresponds to "a processing unit coupled to the input for processing the input signal and the model group to generate a hybrid speech model associated to the given speech element, said hybrid speech model being a combination of speech models in said plurality of speech models on the basis of the input signal derived from the spoken utterance"; and adaptation occurs during recognition with the inherent output of the recognition result from the recognizer (col. 2, lines 45-50), which corresponds to "an output for releasing a signal indicative of said hybrid speech model associated to the given speech element in a format suitable for use by a speech recognition device."

Regarding claim 2, Kuhn teaches everything claimed, as applied above (see claim 1); in addition, Kuhn teaches the modeling of speech units (such as a phrase, word, subword, phoneme or the like) (col. 3, lines 4-7), which corresponds to "the given speech element is an element selected from the group consisting of phones, diphones, syllables and words."

Regarding claim 3, Kuhn teaches everything claimed, as applied above (see claim 2); in addition, Kuhn teaches the training of a speaker dependent model (one for each sound unit) (col. 5, line 30-33), which corresponds to "input signal derived from a spoken utterance is indicative of a speaker specific speech model associated to the at least one speech element."

Regarding claim 4, Kuhn teaches everything claimed, as applied above (see claim 3); in addition, Kuhn teaches the creation of a new speaker dependent model (col. 5, lines 22-57), which corresponds to "hybrid speech model is weighted toward the speaker specific speech model."

Regarding claim 5, Kuhn teaches everything claimed, as applied above (see claim 4); in addition, Kuhn teaches that the speaker dependent model serves to estimate the linear combination of coefficients that will comprise the adapted model **44** (col. 5, 50-57), which corresponds to "said hybrid speech model is derived by computing a linear combination of the speech models in said model group."

Regarding claim 6, Kuhn teaches everything claimed, as applied above (see claim 5). In addition, Kuhn teaches the following: the use of an iterative process where multiple speaker dependent inputs can be used during training (col. 5, lines 22-58, in particular lines 54-58), which corresponds to "a first input and wherein said input signal is a first input signal, said apparatus further comprising: a) a second input for receiving a second input signal conveying a data element identifying the given speech element"; speaker dependent and speaker independent HMMs (models) for each sound unit (col. 4, lines 40-46), which corresponds to "b) a database of model groups comprising a

plurality of model groups, each model group being associated to a respective speech element, each model group comprising a set of speech models"; and the construction of a new model for the for a given sound (in the supervisor mode) (col. 5, lines 22-58, in particular lines 33-36), which corresponds to "said processing unit being further operative for extracting from said database of model groups a certain model group associated to the data element received at said second input identifying the given speech element."

Regarding claim 8, Kuhn discloses an algorithm (Fig. 3) for speaker adaptation inherently implemented with a computational unit that includes the following: a new speaker input (col. 5, lines 26-39, Fig. 3, **40**), which corresponds to "an input for receiving an input signal derived from a spoken utterance that contains at least one speech element that potentially matches the given speech element"; a set of HMM's **44** (one for each sound) inherently stored in a memory (col. 5, lines 23-34), which corresponds to "a memory unit for storing a model group associated to the given speech element, said model group comprising a plurality of speech models, each speech model of said plurality of speech models being a different representation of the given speech element"; with an inherent processing unit to generate an adapted model based in the input (Fig. 3 **52** col. 5, lines 50-57), which corresponds to "a processing unit coupled to the input for processing the input signal and the model group to generate a hybrid speech model associated to the given speech element, said hybrid speech model being a combination of speech models in said plurality of speech models on the basis of the input signal derived from the spoken utterance"; the adaptation of the models during

recognition (col. 2, lines 45-50) with the inherent release of the recognition result, which corresponds to "an output for releasing a signal indicative of said hybrid speech model associated to the given speech element in a format suitable for use by a speech recognition device."

Regarding claim 9, Kuhn discloses a device for speaker adaptation that includes speaker independent and dependent models (col. 2, lines 34-50) which includes the following features: a new speaker input (col. 5, lines 26-39, Fig. 3, **40**), which corresponds to "an input for receiving an input: signal derived from a spoken utterance that contains at least one speech element that potentially matches the given speech element"; two sets of HMMs inherently stored in a memory, one speaker dependent **44** and the other adapted **52** (col. 5, lines 23-34), which corresponds to "a model group associated to the given speech element, said model group comprising a plurality of speech models, each speech model of said plurality of speech models being a different representation of the given speech element, said model group comprising two sets of speech models namely a first set having speech models of a first type and a second set having speech models of a second type, each speech model of a first type in said first set being associated to a speech model of the second type in the second set"; an inherent processing unit for creating a speaker dependent model and a supervector (col. 5, lines 40-49 **44 48**), which corresponds to "a) processing the input signal and the model group to generate a hybrid speech model associated to the given speech element, said hybrid speech model being a combination of speech models of the first type in said plurality of speech models on the basis of the input signal derived from the

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spoken utterance"; the creation of an adapted model **52** from the supervector (col. 5, lines 39-57), which corresponds to "b) processing the hybrid speech model to generate a complex speech model associated to the given speech element, said complex speech model being a combination of speech models of the second type in said plurality of speech models"; and the inherent release of the recognition result while the training is occurring (col. 2, lines 45-50), which corresponds to "an output for releasing a signal indicative of said complex speech model associated to the given speech element in a format suitable for use by a speech recognition device."

Regarding claim 10, Kuhn teaches everything claimed, as applied above (see claim 9); in addition, Kuhn teaches the use of an adapted model that is modified using the speaker dependent model (Fig. 3, col. 5, lines 26-58), which corresponds to "speech model of a second type is indicative of a speech model having a higher complexity than a speech model of a first type to which it is associated."

Regarding claim 11, Kuhn teaches everything claimed, as applied above (see claim 10); in addition, Kuhn teaches that the Hidden Markov Models (used in Kuhn's models) can be used to model speech units such as a phrase, word, subword, phoneme or the like (col. 3, lines 4-7), which corresponds to "speech element is indicative of a data element selected from the set consisting of phones, diphones, syllables and words."

Regarding claim 12, Kuhn teaches everything claimed, as applied above (see claim 11); in addition, Kuhn teaches that the training can be done in a supervisor mode where the training system knows the contents of the training speech in advance (col. 5,



lines 33-36), which corresponds to "said input signal derived from a spoken utterance is indicative of a speaker specific speech model associated to the at least one speech element."

Regarding claim 13, Kuhn teaches everything claimed, as applied above (see claim 12); in addition, Kuhn teaches that the models are modified with speaker specific data (Fig. 3, col. 5, lines 23-57), which corresponds to "said hybrid speech model is weighted toward the speaker specific speech model."

Regarding claim 14, Kuhn teaches everything claimed, as applied above (see claim 13); in addition, Kuhn teaches that the speaker dependent model **44** serves to estimate the linear combination of coefficients that will comprise the adapted model for the new speaker (col. 5, lines 44), which corresponds to "said hybrid speech model is derived by computing a linear combination of the speech models of the first type."

Regarding claim 15, Kuhn teaches everything claimed, as applied above (see claim 14); in addition, Kuhn teaches that the speaker dependent model serves as an estimate of the coefficients that will comprise the adapted model (col. 5, lines 50-67, col. 6, lines 1-12), which corresponds to "first set of parameters indicative of weights associated to speech models of the first type, said complex speech model being derived by computing a second linear combination of the speech models of the second type, said second linear combination being characterized by a second set of parameters indicative of weights associated to speech models of the second type."

Regarding claim 16, Kuhn teaches everything claimed, as applied above (see claim 15); in addition, Kuhn teaches that the speaker dependent model serves as an

estimate for the coefficients that will comprise the adapted model (col. 5, lines 50-67, col. 6, lines 1-12), which corresponds to "said first set of parameters and said second set of parameters are indicative of substantially same weights."

Regarding claim 17, Kuhn teaches everything claimed, as applied above (see claim 10). In addition, Kuhn teaches the following: a supervisor mode indicating that the input signal is known (col. 5, lines 33-35), which corresponds to "a) a second input for receiving a second input signal indicative of a data element identifying the given speech element"; the use of multiple models with corresponding information (Fig. 3, 44 48 42, col. 5, lines 24-58), which corresponds to "b) a database of model groups comprising a plurality of model groups, each model group being associated to a respective speech element, each model group comprising two sets of speech models namely a first set having speech models of a first type and a second set having speech models of a second type, each speech model of a first type in said first set being associated to a speech model of the second type in the second set"; the creation of a new supervector with the iterative process used to construct another set of HMMs (inherent selectivity associated with a specific speech element if in the supervised mode) (col. 5, lines 22-58), which corresponds to "unit being further operative for extracting from said database of model groups a certain model group associated to the data element received at said second input identifying the given speech element."

Regarding claim 18, Kuhn teaches techniques and algorithms for speaker adaptation based on eigenvoices using multiple model groups (Fig. 3, 44 48 52) where the model groups contain HMMs that can be used to represent phrases, words,

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subwords, or phonemes (col. 3, lines 4-8) and inherently implemented on a computational device, which corresponds to "a data structure for storing a plurality of model groups, each model group being associated to a respective speech element in a phonetic alphabet, each model group comprising a plurality of speech models, each model group being suitable for use by a processing device."

Regarding claim 19, Kuhn teaches everything claimed, as applied above (see claim 18); in addition, Kuhn teaches that the speech units can be phrases, words, subwords or phonemes or the like (col. 3, lines 4-7), which corresponds to "the given speech element is indicative of a data element selected from the set consisting of phones, diphones, syllables and words."

Regarding claim 20, Kuhn teaches everything claimed, as applied above (see claim 19); in addition, Kuhn teaches the use of a data structure with multiple model groups with the adapted model being the most highly processed (Fig. 3, **44 48 52**, col. 5, lines 23-58), which corresponds to "wherein each model group comprises two sets of speech models namely a first set having a plurality of speech models of a first type and a second set having a plurality of speech models of a second type, each speech model of a first type in said first set being associated to a speech model of the second type in the second set, each speech model of the second type being indicative of a speech model having a higher complexity than a speech model of the first type to which the speech model of the second type is associated."

Regarding claim 21 and 36, Kuhn discloses an adaptable speech recognition system (col. 2, lines 45-50) with the following features: a speaker input (Fig. 3 **40**) which

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when in supervised mode knows the content of an input in advance (col. 5, lines 33-36), which corresponds to "an input for receiving an input signal indicative of a spoken utterance that is indicative of at least one speech element"; procedures for training of a speaker dependent recognizer (Fig. 3 **42 44**, lines 39-49), which corresponds to "a first processing unit coupled to said input operative for processing the input signal to derive from a speech recognition dictionary at least one speech model associated to a given speech element that constitutes a potential match to the at least one speech element"; procedures to construct a supervector (Fig. 3 **46 48 38**), which corresponds to "a second processing unit coupled to said first processing unit for generating a modified version of the at least one speech model or the basis of the input signal"; and procedures to construct a new set of HMMs based on the supervector (Fig. 3, **50 52**), which corresponds to "a third processing unit coupled to said second processing unit for processing the input signal on the basis of the modified version of the at least one speech model to generate a recognition result indicative of whether the modified version of the at least one speech model constitutes a match to the input signal"; and since Kuhn's system is a recognition system that automatically adapts during recognition (col. 2, lines 45-50), Kuhn's system inherently releases recognition results, which corresponds to "an output for releasing a signal indicative of the recognition result."

Regarding claim 22, Kuhn teaches everything claimed, as applied above (see claim 21); in addition, Kuhn teaches the processing of the input to produce a speaker dependent model of a specific input when in the supervised mode (Fig. 3 **40 42 44**, col. 5, 39-49), which corresponds to "wherein said first processing unit is operative for

generating a speaker specific speech model derived on the basis of the input signal, the speaker specific speech model being indicative of the acoustic characteristics of the least one speech element."

Regarding claim 23, Kuhn teaches everything claimed, as applied above (see claim 22); in addition Kuhn teaches that the speaker dependent model **44** serves to estimate the linear combination of coefficients that will comprise the adapted model (col. 5, lines 50-55), which corresponds to "said modified version of the at least one speech model is indicative of a hybrid speech model associated to the given speech element."

Regarding claim 24, Kuhn teaches everything claimed, as applied above (see claim 23). In addition, Kuhn teaches the following: the transfer of data between the procedures (indicated by the arrows in Fig. 3 between elements **42 44** and **46**), which corresponds to "coupling member for allowing data exchange between the first processing unit and the second processing unit, said coupling member being suitable for receiving the speaker specific speech model derived from the input signal"; a model group containing HMMs with models of speech sounds (Fig. 3, col. 5, lines 26-39), which corresponds to "a model group associated to the given speech element, said model group comprising a plurality of speech models, each speech model of said plurality of speech models being a different representation of the given speech element"; a procedure to construct a new set of HMMs based on the supervector (Fig. 3 **50**), which corresponds to "a functional unit coupled to the coupling member for processing the speaker specific speech model and the model group to generate the hybrid speech model associated to the given speech element, said hybrid speech model

being a combination of speech models in said plurality of speech models on the basis of the speaker specific speech model"; and since Kuhn discloses a speech recognition system (col. 2, lines 45-50) it has an inherent means for indicating a particular recognition event on output, which corresponds to "an output coupling member for allowing data exchange between the second processing unit and the third processing unit, said output coupling member being suitable for releasing a signal indicative of the hybrid speech model associated to the given speech element."

Regarding claim 25, Kuhn teaches everything claimed, as applied above (see claim 24); in addition, Kuhn teaches that the dependent model **44** serves to estimate the linear combination of coefficients that will comprise the adapted model (col. 5, lines 50-54), which corresponds to "said hybrid speech model is weighted toward the speaker specific speech model."

Regarding claim 26, Kuhn teaches everything claimed, as applied above (see claim 24); in addition, Kuhn teaches that the dependent model **44** serves to estimate the linear combination of coefficients that will comprise the adapted model (col. 5, lines 50-54), which corresponds to "said hybrid speech model is derived by computing a linear combination of the speech models in said group of speech models."

Regarding claim 27, Kuhn teaches everything claimed, as applied above (see claim 24); in addition, Kuhn teaches the following: transfer of data between the speaker dependent building portion of the model and the adapted model (indicated by arrows in Fig. 3), which corresponds to "a) a second coupling member for allowing data exchange between the first processing unit and the second processing unit, said second coupling

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member being suitable for receiving a data element identifying the given speech element"; groups of models (Fig. 3 **44 48 52**) containing HMMs of speech sounds (col. 3, lines 3-7, col. 5, lines 50-57), which corresponds to "b) a database of model groups comprising a plurality of model groups, each model group being associated to a respective speech element, each model group comprising a set of speech models"; a procedure **50** for constructing a new set of HMMs based on the supervector **48** that can access the models with the adapted model (Fig. 3 **50 52**), which corresponds to "functional unit being further operative for extracting from said database of model groups a certain model group associated to the data element received at said second coupling member identifying the given speech element."

Regarding claim 28, Kuhn teaches everything claimed, as applied above (see claim 22); in addition, Kuhn teaches a HMM contained in the adapted model **52** is adapted to the speech input and can be constructed by an iterative process (col. 5, lines 54-56), which corresponds to "said modified version of the at least one speech model is indicative of a complex speech model associated to the given speech element."

Regarding claim 29, Kuhn teaches everything claimed, as applied above (see claim 28). In addition, Kuhn teaches the following: the speaker specific model **44** generated from the input is coupled (arrow between **44** and **46** in Fig. 3) to the next stage for further processing, which corresponds to "coupling member for allowing data exchange between the first processing unit and the second processing unit, said coupling member being suitable for receiving the speaker specific speech model derived from the input signal"; a speaker specific model group **44** containing multiple

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HMMs for sounds and an adapted model group **52** with corresponding sounds (col. 5, lines 22-57), which corresponds to "a model group associated to the given speech element, said model group comprising a plurality of speech models, each speech model of said plurality of speech models being a different representation of the given speech element, said model group comprising two sets of speech models namely a first set having speech models of a first type and a second set having speech models of a second type, each speech model of a first type in said first set being associated to a speech model of the second type in the second set"; a procedure for training speaker dependent models (Fig. 3 **42**) and an adapted model **52** that combines speaker specific and speaker independent data (col. 5, lines 39-58), which corresponds to "a) processing the speaker specific speech model and the model group to generate a hybrid speech model associated to the given speech element, said hybrid speech model being a combination of speech models of the first type in said plurality of speech models on the basis of the speaker specific speech model"; a procedure to construct new sets of HMMs **52** based on a supervector (Fig. 3 **50**) where the resulting models are a combination of the speaker independent and speaker dependent data (col. 5, 22-58), which corresponds to "b) processing the hybrid speech model to generate a complex speech model associated to the given speech element, said complex speech model being a combination of speech models of the second type in said plurality of speech models"; and since Kuhn discloses a speech recognition system (col. 2, lines 45-50) it has an inherent means for indicating a particular recognition event on output, which corresponds to "output coupling member for allowing data exchange between the



second processing unit and the third processing unit, said coupling member being suitable for releasing a signal indicative of the complex speech model associated to the given speech element."

Regarding claim 30, Kuhn teaches everything claimed, as applied above (see claim 29); in addition, Kuhn teaches that the adapted model is based on a linear combination of the components from the speaker dependent model which might entail multiple iterations (col. 5, lines 39-65), which corresponds to "any speech model of the second type is indicative of a speech model having a higher complexity than a speech model of the first type to which it is associated."

Regarding claim 31, Kuhn teaches everything claimed, as applied above (see claim 30); in addition, Kuhn teaches that the adapted model is a linear combination of the coefficients from the speaker dependent model (col. 5, lines 50-55), which corresponds to "wherein said hybrid speech model is weighted toward the speaker specific speech model."

Regarding claim 32, Kuhn teaches everything claimed, as applied above (see claim 31); in addition, Kuhn teaches that the adapted model is a linear combination of the coefficients from the speaker dependent model (col. 5, lines 50-55), which corresponds to "said hybrid speech model is derived by computing a linear combination of tire speech models of the first type."

Regarding claim 33, Kuhn teaches everything claimed, as applied above (see claim 32). In addition, Kuhn teaches the use of an iterative procedure where to construct a new supervector from the adapted model and there after to construct

another set of HMMs from which a further adapted model may be constructed (col. 5, lines 50-57), which corresponds to "wherein said linear combination is a first linear combination and is characterized by a first set of parameters indicative of weights associated to speech models of the first type, said complex speech model being derived by computing a second linear combination of the speech models of the second type, said second linear combination being characterized by a second set of parameters indicative of weights associated to speech models of the second type."

Regarding claim 34, Kuhn teaches everything claimed, as applied above (see claim 33); in addition, Kuhn teaches that the constructing of supervector **48** may be accomplished through a computationally simple projection operation or the like (col. 5, lines 59-67), which corresponds to "said first set of parameters and said second set of parameters is indicative of substantially the same weights."

#### ***Citation of Pertinent Art***

1. The following prior art made of record but not relied upon is considered pertinent to the applicant's disclosure:

- a) Kuhn et al. (U.S. Patent 6,343,267) discloses methods for dimensionality reduction for speaker normalization and speaker and environment adaptation using eigenvoice techniques.
- b) Newman et al. (U.S. Patent 6,151,575) uses a linear relationship between an initial model and a source-adapted model.

- c) Nguyen et al. (U.S. Patent 6,263,309) uses a maximum likelihood method for finding an adapted speaker model in eigenvoice space.
- d) Hackbarth et al. (U.S. Patent 5,170,432) discloses a method for adaptive speaker recognition using hypotheses generated based in extracted feature vectors from a speech signal.
- e) Huo et al., (Proceedings of ICSLP 2000 Vol. IV, pp. 480-438) teaches an adaptation technique that requires relatively few parameters
- f) Westwood ("Speaker Adaptation Using Eigenvoices," Mphil in Computer Speech and Language Processing, Dept. of Engineering, Cambridge University, August 31, 1999) gives an overview of adaptive techniques using eigenvoices.

### ***Conclusion***

Any response to this office action should be mailed to:

Commissioner of Patents and Trademarks  
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or faxed to:

(703) 872-9314

Hand-delivered responses should be brought to:

Crystal Park II  
2121 Crystal Drive  
Arlington, VA.  
Sixth Floor (Receptionist)

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Dr. V. Paul Harper whose telephone number is (703) 305-4197. The examiner can normally be reached on Monday through Friday from 8:00 a.m. to 4:30 p.m.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Marsha D. Banks-Harold, can be reached on (703) 305-4379. The fax phone number for the Technology Center 2600 is (703) 872-9314.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the Technology Center 2600 Customer Service office whose telephone number is (703) 306-0377.

VPH/vph  
October 24, 2002

*Marsha D. Banks-Harold*  
MARSHA D. BANKS-HAROLD  
SUPERVISORY PATENT EXAMINER  
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